EXHIBIT A

Genuity

IP Telecom Engineering

VoIP RTP Routing

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Abstract

This document describes a method for raising the quality of Voice over IP calls by controlling the path of the RTP media stream to ensure that where possible the Voice media stream traverses a managed IP network and avoids unmanages networks, or public peering points.

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Abstract

This invention disclosure describes a method for controlling call signaling and media (voice) flow between a originating and terminating VoIP endpoint such that packets carrying the media in a VoIP call are forced through network elements of a given IP address. This in turn provides a network service provider or carrier a method to:

- increase the quality of Voice over IP calls and
- prevent call originator and/or call terminators from bypassing the network service provider or carrier's network.

In the first case, the quality of a VoIP call can be insured by controlling the path of the media stream to ensure that these packets traverses a known (and presemably managed IP network). In this case, the path of the voice packets can avoid congested networks or peering points. For VoIP calls, the media is typically carried by the RTP protocol.

In the second case, since packets corresponding to voice calls in both directions can be forced through specific network elements, network address translation (NAT) can be used to hide the terminating VoIP gateway's address from the call originator, and similarly hide the originator's VoIP gateway address from the call terminator. This process - called "blind re-file" - allows network providers to use networks other than their own to terminate calls without revieling this fact (or any network details) to either call orginator or call terminator. This prevents these network users or carriers from bypassing this "facilitor" network.

The overall idea for controlling RTP routing combines the ideas of a call signaling proxy and a call media proxy server. The signaling proxy receives call signaling information from a originating voip endpoint and relays call signaling information from that endpoint to the correct destination endpoint (or to another signaling proxy that will relay the call signaling information to the destination endpoint.) The media proxy server has a list of static or dynamic virtual IP addresses that represent media endpoints, gateways, other media proxies. These virtual addresses are used by the call signaling proxy to direct the originating media endpoint to use the RTP media proxy service once it has determined which endpoint will handle the call. The originating voip endpoint then streams it's media to the media proxy. In the case of blind refile, the media proxy and the signaling proxy use the well know Network Address Translation method [reference] to replace their own IP adress with the address of the "next-hop."

Originating voip endpoints can select a call signaling and media proxies that will provide the best quality of service for their calls by testing the quality of the network connection from their point of presence to each of the call signaling and media proxies. This in essance provides them with the best path to the managed network.

Background

PC to phone VoIP calling providers are all looking for ways to differentiate their service from that of the competition and or created differentiated levels of service within their own product suite. Likewise phone-to-phone VoIP providers are looking for improved quality of service and higher levels of assured quality. VoIP carriers are looking for a way to distinguish their VoIP termination or origination services from that of their competitors.

Different VoIP carriers have configured their networks in different ways. Some providers use Internet connections in each of their locations to connect their points of presence. Others carriers have multiple Internet connections at each Pop and switch between these connections as the quality of each network connection changes. Still other carriers have built private networks to control the quality of voice on their networks, and/or have adapted a model where a private network connects all of their POPS together, but allows each POP to be able to receive traffic from other points on the Internet.

In the IP phone to PSTN model one endpoint is always connected to the Internet. This endpoint is either an end user pc or IP telephone. The call signaling proxy allows the IP endpoint vendors and providers to build the intelligence into their clients to find the closest interconnection point to each of their VoIP carriers. The call signaling and media proxy allows carriers to control how the RTP stream traverses their network. It also allows carriers to control the path of the RTP stream when they hand off VoIP traffic to other VoIP carriers. It also allows carriers to build out strategic connections from their proxy locations to points within ISP networks where there are large concentrations of pc to phone users. It also allows them to control the RTP and signaling traffic coming from customers directly connected to their VoIP networks when they are using another ITSP or carrier to terminate this traffic.

In addition to differing network topologies, VoIP carriers typically cannot terminate calls directly to all (globally) local PSTN carriers. For example, VoIP carrier A may be able to terminate calls to Local Exchange Carriers (LECs) in the United States, but may not have direct connections to LECs in China, while Carrier B may have limited capabilities in the United States, but may have an extensive network in the Far East. However, Carrier A may wish to terminate calls in China without revealing to its customers that it must use Carrier B's network to do so. (i.e., Carrier A wishes to prevent its customers from bypassing it and terminating calls directly to Carrier B.) In this case, by forcing both signaling and media packets to flow through a given fixed point on Carrier A's network, and using NAT, Carrier A can hide Carrier B's network from its customers, and the reverse.

Terminology and Definitions

Service Provider – VoIP network provider who provides wholesale VoIP network servives to other providers. The provider could be an Internet telephony service provider (ITSP). VoIP network operator.
Customer – Retail VoIP provider who provides services to end users. The retail VoIP provider uses the services of of service providers as described above.
End User – Retail client who uses the services of a retail VoIP provider. Assumption for many of these cases is that the end user is connected to the internet using an undetermined Internet service provider (ISP).
PSTN – Public Switch Telephone Network
VoIP – Voice over Internet Protocol
RTP – Real Time Protocol – This is the protocol used to transport voice samples within VoIP networks
Call Signaling – SIP, H323 MGCP or other call control and signaling protocols
SIP – Session Initiation Protocol
MGCP – Media Gateway control Protocol
ITSP – Internet Telephony Service Provider
VoIP Carrier – Telephony provider who uses an IP network to transport their voice traffic
PC to Phone VoIP calling providers – Retailers of VoIP PC to phone calls
Pop – Point of Presence
PC – Personal computer

Call scenarios

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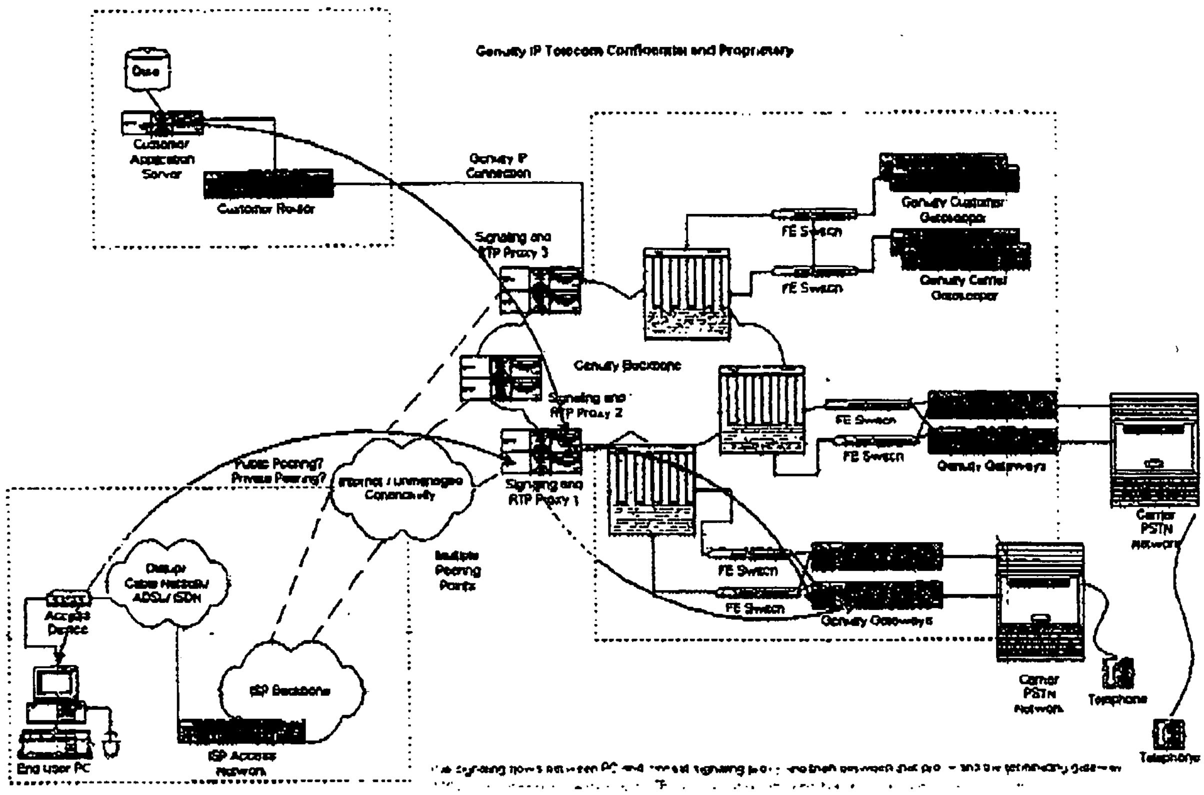
PC to phone signaling and media proxy – Topology, Signaling and Media flows



The call signaling and media proxy allows voice media traffic between a VoIP PC client and a wholesale internet telephony service provider VoIP gateway to remain on the wholesale internet telephony service provider's network for the longest possible portion of its travel. The proxy also allows traffic of directly connected gateway customers to remain on the wholesale internet telephony service provider VoIP network even if the traffic is destine for a termination partner

Customers of wholesale Internet telephony service provider's can implement a process within their VoIP clients to detect the closest RTP media proxy to that individual client. This process could use a series of pings, trace routes, or other messages to each of the media server to judge which represented the closest, or shortest path, or which had the most reliable connection. This is necessary because it is important to determine the shortest path between the end user pc client and the wholesale Internet telephony service provider network, not the customer server and the wholesale Internet telephony service provider network.

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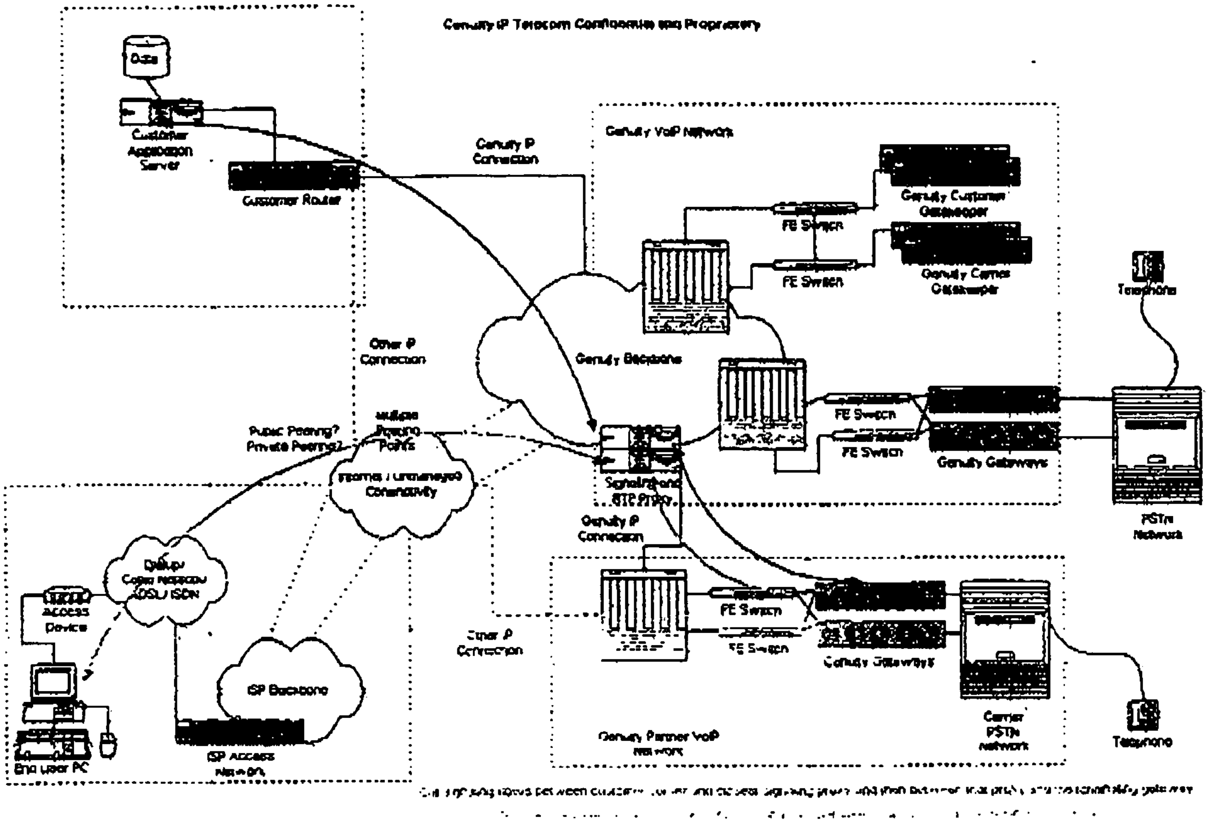
The customer's server can then setup a call using the signaling proxy associated. The signaling proxy can be programmed to use a media proxy or set of media proxies that have been selected as closest to the customer. This signaling proxy also acts as an entrance point into the service provider's least cost routing mechanism. The service provider can route calls across its network, as it normally would taking advantage of all routing mechanisms normally used. Once a destination gateway is determined for the call the signaling proxy sets up a call to the destination gateway. The service provider call signaling proxy will instruct the customer's server to send the client media stream to a particular IP address and port associated with the media proxy. The service provider signaling proxy also instructs the destination gateway to send it's media stream to a particular IP address and port associated with the media proxy. The Proxy then uses network address translation (NAT) or a similar mechanism to change the destination IP addresses on the RTP packets that travel between the destination gateway and the pc client. This functionality requires the signaling proxy and the media proxy to have an interface where the signaling proxy can define virtual sets of IP addresses and ports assigned to the destination gateways and to the pc clients. This would need to happen in real time as the actual RTP address of both the gateway and the PC client are made available only during the call setup process.

Carrier signaling and media proxy – Topology, Signaling and Media flows



Service providers can also use then proxy as a way to control how calls are sent to their partners who terminate calls for them using VoIP interconnections. This allows Internet telephony service providers to manage the quality of calls that may not terminate on equipment that they own by being able to direct RTP traffic to locations on their network and then direct the RTP streams to their termination VoIP partners. Without this media proxy the RTP for these calls would traverse unmanaged IP connections.

Drawing 3 describes a RTP flow from a pc client to a termination partner can be directed through an Internet telephony service providers media and signaling proxy. A pair of proxies could also be used to control the entry point into the Internet telephony service providers network and the exit from the providers network to their partners point of presence.

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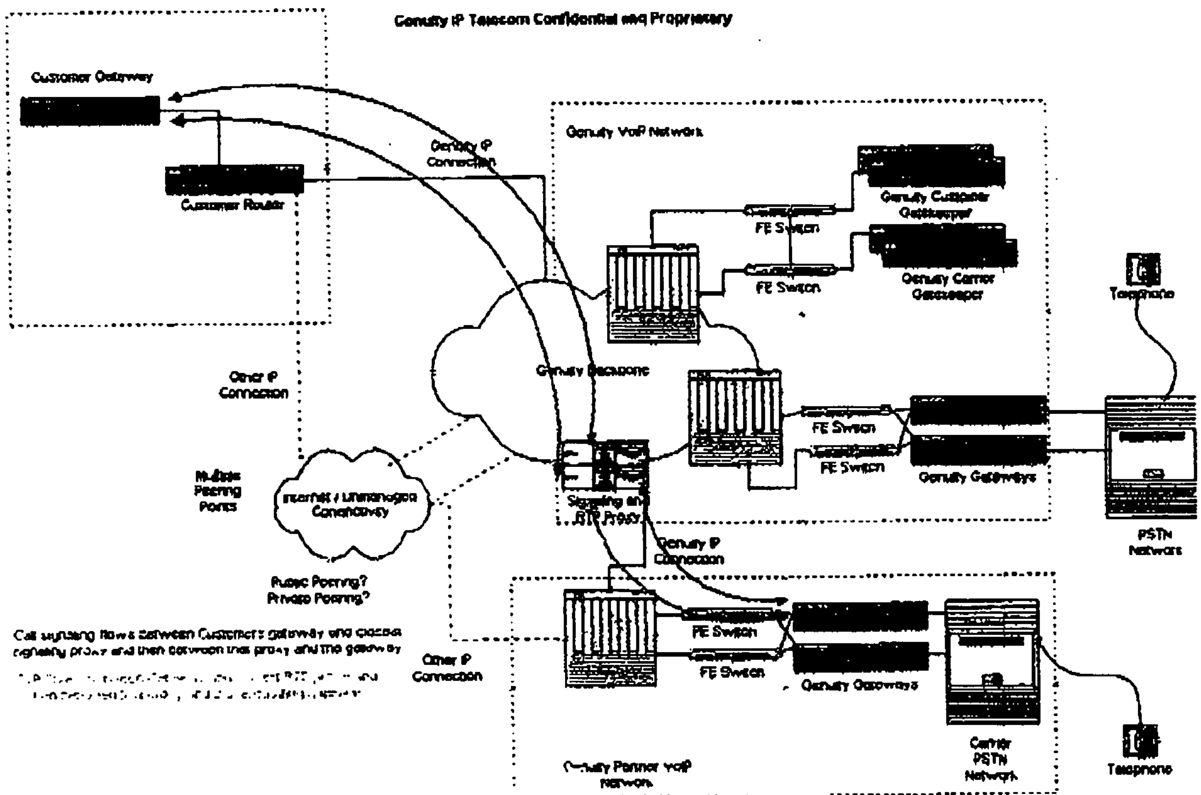
IP connected customer with carrier proxy – Topology, Signaling and Media flows



Customers who purchase direct connections to Internet telephony service provider backbones expect a higher quality of service than pc to phone calling or customer using open Internet connections to transport their VoIP call signaling and media. When a provider chooses to send calls to one of their IP connected partners for termination there is a desire to ensure that customers RTP is handled as if the gateway was on the service providers network and not on a network of a partner. Because both the customer and the terminating partner may have other internet connections that are unmanaged it is necessary to use a proxy to route the RTP for calls provided by the service provider over their managed network. Directly connected customers with fixed locations can be routed directly to gateway owned by the service provider or to proxies that serve individual terminating partners. For this configuration it is ideal to locate the proxy on the service providers network close to the interconnection point with the terminating partner.

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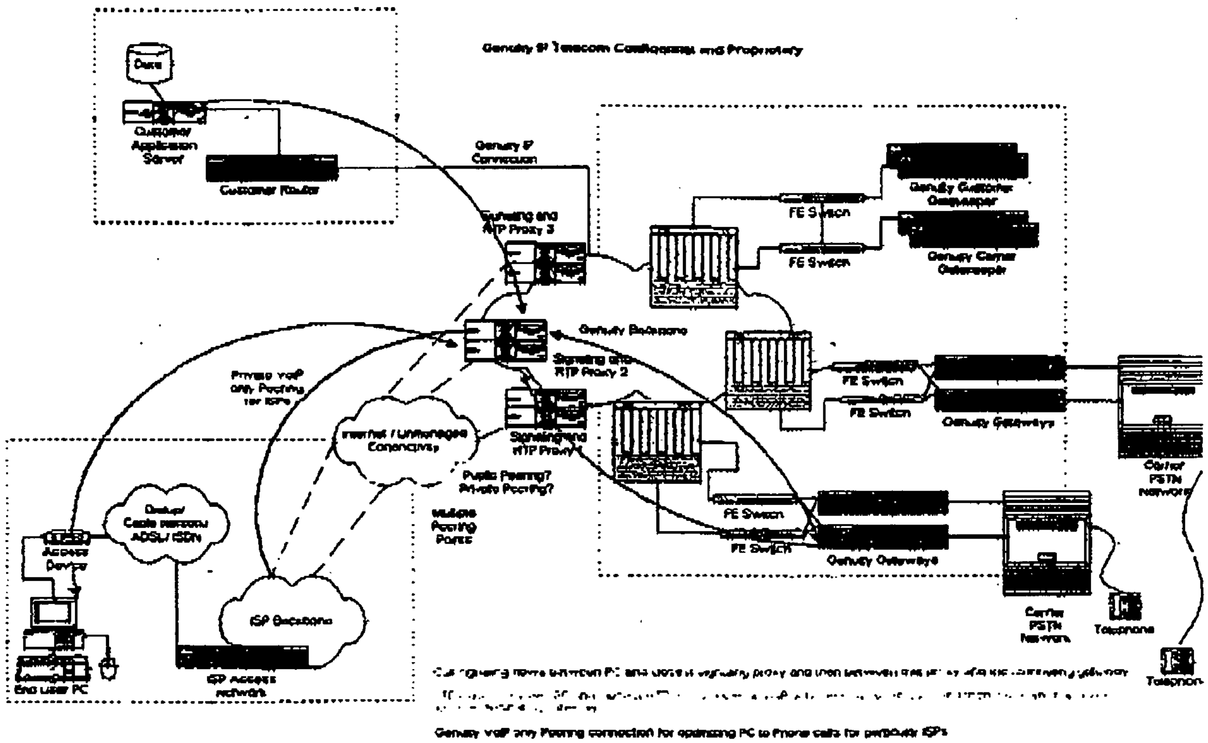
ISP private VoIP peering with Proxy – Topology, Signaling and Media flows



ITSP carriers or their PC to phone customers who use ITSP networks for transport and termination may want to improve the voice quality for customers using certain ISPs for connectivity while using their PC to phone VoIP service. An ISP may also be a PC to phone provider and want to better connect their customers to the VoIP network. Traditional peering arrangements provide a means to connect ISP networks to ITSP networks, but traditionally are not for VoIP traffic only. By setting up a call signaling and RTP proxy that has direct connections to an ISP network peering can be setup for only VoIP traffic. When a ITSP and a ISP peer and only advertise the IP addresses of a group of call signaling and RTP proxies to the ISPs network this connection can be used for only VoIP traffic. In this way the VoIP call signaling and media flows will avoid the ISPs normal peering connections and use the dedicated VoIP peering connection. This allows the ISP and ITSP to better manage the PC to Phone traffic and quality of service. PC to phone services that employ this type of interconnection can demonstrate an improved quality of service differentiating their calls from calls routed over normal peering connections.